

What Is Claimed Is:

sub B1
2r

1 1. A method for reducing overhead and latency and handling packet
2 loss in a voice and data over Internet Protocol (VoIP) data packet, transmitted
3 between originating and destination gateways in an Internet telephony system,
4 comprising the steps of:

- 5 (1) compressing data streams from a plurality of concurrent
6 calls from a plurality of channels into packets;
7 (2) aggregating said packets into the larger data packet;
8 (3) transmitting the data packet between the originating and
9 destination gateways through a single virtual connection; and
10 (4) controlling the transmission of the data packet between the
11 originating and destination gateways by defining the format for the data packet,
12 and updating and synchronizing the header information in the data packet.

1 2. The method of claim 1, wherein step (2) further comprises the
2 step of aggregating said packets into the data packet, wherein the data packet
3 comprises a plurality of data frames and a plurality of header frames,
4 comprising at least one header frame selected from the group consisting of a
5 time stamp header, local network address header, IP address header and UDP
6 header and at least two header frames selected from the group consisting of a
7 version number header, channel present header, channel header and control
8 information header.

1 3. The method of claim 1, wherein step (1) further comprises the step
2 of converting analog data streams to digital data streams prior to compressing
3 said data streams into said packets.

1 4. The method of claim 1, further comprising the step of transmitting
2 a check sequence data packet at regular packet intervals, configurable to

tradeoff between increased tolerance to loss and bandwidth, wherein a parity system and the information located inside of said check sequence data packet is used to regenerate missing or damaged data in the previously transmitted data packet.

5. A method for regenerating missing or damaged data in a data packet transmitted in an Internet telephony system, comprising the steps of:

(1) transmitting a check sequence data packet after the transmission of every third data packet, wherein information located inside of said check sequence data packet is used to regenerate the missing or damaged data; and

(3) using a parity system to regenerate the missing or damaged data.

6. A system for reducing overhead and latency and handling packet loss in a voice and data over Internet Protocol (VoIP) data packet, transmitted over a UDP/IP connectionless protocol between originating and destination gateways, said system comprising:

media framing means for aggregating packets from a plurality of concurrent calls from a plurality of channels into the larger data packet;

transmission control means for defining the format for the data packet, and updating and synchronizing header information in the data packet;

redundancy means for regenerating missing or damaged data in the data packet; and

a single virtual connecting means for transmitting the data packet from the originating gateway to the destination gateway.

7. The system of claim 6, wherein the data packet comprises a plurality of data frames and a plurality of header frames, comprising at least one header frame selected from the group consisting of a time stamp header,

4 local network address header, IP address header and UDP header and at least
5 two header frames selected from the group consisting of a version number
6 header, channel present header, channel header and control information header.

1 8. The system of claim 6, further comprising:
2 means for transmitting and receiving data streams;
3 means for converting analog data streams to digital data streams;
4 means for compressing digital data streams into said packets; and
5 means for transmitting a check sequence data packet after the
6 transmission of every third data packet.

1 9. The system of claim 8, wherein said check sequence data packet
2 is formatted to regenerate said missing or damaged data with information located
3 inside of said check sequence data packet, and use a parity system to regenerate
4 said missing or damaged data.

1 10. An Internet telephony system for regenerating missing or
2 damaged data in a data packet, comprising:
3 redundancy means for transmitting a check sequence data packet
4 after every three or more data packets; and
5 means for regenerating the missing or damaged data with the
6 information located inside of said check sequence data packet.

1 11. The system of claim 10, further comprising means for
2 implementing a parity system to regenerate said missing or damaged data.

1 12. A computer program product comprising a computer useable
2 medium having computer program logic recorded thereon for enabling originating
3 and destination gateways to transmit or receive data streams or data packets in an

Internet telephony system and for reducing VoIP packet overhead and latency and handling packet loss, said computer program logic comprising:

a first computer program product means for compressing the data streams from a plurality of concurrent calls from a plurality of channels into packets;

a second computer program product means for aggregating said packets into the larger data packets;

a third computer program product means for transmitting the data packets between the originating and destination gateways through a single virtual connection;

a fourth computer program product means for controlling the transmission of the data packets between the originating and destination gateways by defining the format for the data packets, and updating and synchronizing header information in the data packets; and

a fifth computer program product means for determining if the data packets contain missing or damaged data and regenerating said missing or damaged data in the data packets.

13. The computer program product of claim 12, wherein said second computer program product means further comprises computer program product means for aggregating packets into the data packets comprising a plurality of data frames and a plurality of header frames, wherein said header frames comprises at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least two header frames selected from the group consisting of a version number header, channel present header, channel header and control information header.

14. The computer program product of claim 12, wherein said first computer program product means further comprises computer program product

3 means for converting analog data streams to digital data streams prior to
4 compressing the data streams into said packets.

1 15. The computer program product of claim 14, wherein said fifth
2 computer program product means further comprises computer program product
3 means for transmitting a check sequence data packet after every three data
4 packets and using a parity system and the information located inside of said check
5 sequence data packet to regenerate said missing or damaged data.

1 16. A computer program product comprising a computer useable
2 medium having computer program logic recorded thereon for enabling originating
3 and destination gateways to transmit or receive data streams or data packets in an
4 Internet telephony system and for regenerating missing or damaged data in the
5 data packet, comprising:

6 a first computer program product means for transmitting a check
7 sequence data packet at regular packet intervals, configurable to tradeoff
8 between increased tolerance to loss and bandwidth; and

9 a second computer program product means for regenerating the
10 missing or damaged data by using information located inside of said check
11 sequence data packet.

1 17. The computer program product of claim 26, further comprising
2 a third computer program product means for using a parity system to regenerate
3 the missing or damaged data.